CLAIMS:

1. A method wherein multiple input signals are subjected to a combination process of adaptive beamforming and adaptive echo cancelling, characterized in that for each of the input signals an individual processing history of adaptive echo cancelling data is kept and combined with current adaptive beamforming data.

2. The method according to claim 1, characterized in that the combined adaptive processing is devised such that each of the respective input signals is running through a parallel path containing an acoustic path and a beamformer path, whereafter signals in the parallel paths are summed and processed.

3. The method according to claim 1 or 2, characterized in that adaptive beamforming concerns filtering or weighting of the input signals.

4. An audio processing device comprising at least one parallel acoustic path for providing respective input signals, the acoustic paths are connected in series to beamformer paths, the device comprises an adaptive beamformer and an adaptive echo canceller, characterized in that the adaptive echo canceller is provided with storage means for storing in relation to every input signal, individual processing histories of adaptive echo cancelling data for combination with current adaptive beamforming data.

5. The audio processing device according to claim 4, characterized in that the audio processing device is devised such that each of the respective input signals is running through a parallel path containing an acoustic path and a beamformer path, whereafter signals in the parallel paths are summed and processed.

6. The audio processing device according to claim 4 or 5, characterized in that the adaptive beamformer is a filtered and/or weighted beamformer.

7. The audio processing device according to one of the claims 4-6, characterized in that the adaptive echo canceller comprises a Transform Domain Adaptive Filter, such as for example a Time Domain Adaptive Filter (TDAF), or a Frequency Domain Adaptive Filter (FDAF).

8. The audio processing device according to one of the claims 4-7, characterized in that the adaptive filter comprises a first section for calculating at least one loudspeaker input spectrum and a part of normalized update data, and a second section for performing convolution and calculating echo cancelling coefficient update data.

9. The audio processing device according to claim 8, characterized in that the second adaptive echo canceller section comprises an adaptive summing filter having an input for receiving beamformer filtering or weighting coefficients, the summing filter comprising the storage means for storing in relation to every input signal, individual processing histories of adaptive echo cancelling data for combination with current adaptive beamforming data.

10. A communication device such as found in audio broadcast systems, audio and/or video conferencing systems, speech enhancement, such as in telephone, like mobile telephone systems, speech recognition systems, speaker authentication systems, speech coders and the like, the communication device having an audio processing device according to one of the claims 4-9, the audio processing device comprising at least one loudspeaker, multiple microphones for providing respective inputs signals, which microphones are coupled to the at least one loudspeaker through acoustic paths, an adaptive beamformer and an adaptive echo canceller, characterized in that the adaptive echo canceller is provided with storage means for storing in relation to every input signal an individual processing history of adaptive echo cancelling data for combination with current adaptive beamforming data.